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Review Article

To Investigate the Effect of Oversampling in Reduction of Approximation Error in FIR Filter Design

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Abstract: FIR filters can be designed to have linear phase response and easy to implement. Due to non-recursive nature of FIR filters, they offer significant computational advantages as compared to IIR Filter. In this paper a Gaussian pulse- shaping FIR filter has been designed and influence of its parameter variation has been investigated.

Keywords: FIR Filter, Oversampling, Approximation Error.

1. INTRODUCTION

Finite Impulse Response (FIR) filters are generally chosen for applications where linear phase is important and a decent amount of memory and computational performance are available. They have a widely deployed in audio and biomedical signal enhancement applications. Their all-zero structure ensures that they never become unstable for any type of input signal, which gives them a distinct advantage over the IIR.

Advantages

- *F*IRs can be easily designed to have linear phase. This means that no phase distortion is introduced into the signal to be filtered, as all frequencies are shifted in time by the same amount – thus maintaining their relative harmonic relationships (i.e. constant group and phase delay). This is certainly not case with IIR filters that have a non-linear phase characteristic.
- As FIRs do not use previous output values to compute their present output, i.e. they have no feedback, they can never become unstable for any type of input signal, which is gives them a distinct advantage over IIR filters.
- The Parks-McClellan and ASN Filter Script’s function allow for the design of an FIR with an arbitrary magnitude response. This means that an FIR can be customized more easily than an IIR.
- Fixed point performance: the effects of quantization are less severe than that of an IIR.

2.1 FIR Filters

FIR filters can be designed to have linear phase response and easy to implement. Due to non-recursive nature of FIR filters they offer significant computational advantages as compared to IIR. FIR filters suffer less from the effects of finite word length than IIR filters. FIR filters should be used whenever there is requirement to exploit any of the advantages above, in particular the advantage of linear phase. Specifications of the FIR filter contain to maintain the maximum pass band ripple, maximum stop band ripple, pass band edge frequency and stop band edge frequency. For the calculations of FIR filter coefficients requires large amount of computations. The coefficients can be calculated by using different software and tools such as FDA analysis tool in MATLAB. Filter design and analysis tool (FDA) is one of the most important tools of MATLAB, which is used to design the digital filter blocks more accurate and fast [2].

The most important property to design FIR filter is that its phase response should be linear. For this reason we shall look more closely at this property. When we have to pass any signal through the filter we see the changing in its amplitude and phase. The nature and amount of the variation of the signal is dependent on the amplitude and phase characteristics. We have modified the characteristics of the phase of filter by knowing its group delay. If a signal consists of several frequency components (such as speech waveform or a modulated signal) the phase delay of the filter is the amount of time delay each frequency component of the signal undergoes through the filter. The group delay on the other hand is the average time delay of the composite signal suffers at each frequency. Mathematically the phase delay is the negative of the phase angle divided by frequency is shown in equation (1) and the group delay is the negative of the derivative of the phase with respect to frequency is shown in equation (2).

$$T_p = -\theta(\omega)/\omega \qquad (1)$$

$$T_g = -d\theta(\omega)/d\omega \qquad (2)$$

A filter is said to have a linear phase response if its phase response satisfies one of the following relationships

$$\theta(\omega) = -\alpha\omega \qquad (3)$$

$$\theta(\omega) = \beta - \alpha\omega \qquad (4)$$

If a filter satisfies the condition given in the above equation it will have both constant group and constant phase delay responses. It can be show that from the equation (5) to be satisfied the impulse response of the filter must have positive symmetry and α and β are constants.

$$h(n) = h(N-n-1) \qquad (5)$$

When the condition given in equation 3.6 satisfied the filter will have a constant group delay only. In this case we have the negative symmetry of the filter $0,1,\ldots\left(\frac{N-1}{2}\right)$ where N is odd and $n = 0,1,\ldots\left(\frac{N}{2}\right)-1$, where N is even [5].

$$\alpha = (N-1)/2 \qquad (6)$$

A nonlinear phase characteristic filter will cause a phase distortion in the signal that passes through it. This is because the frequency components in the signal will each be delayed by an amount not proportional to frequency thereby altering their harmonic relationships. Such a distortion is undesirable in many applications, for example music, data transmission, video and biomedicine and can be avoided by using filters with linear phase characteristics over the different frequency bands. However linear filters have low performance during the existence of noise, which is not preservative as well as in problem where system nonlinear information is encountered. For example in image processing applications linear filters are used to blur the edges but cannot eliminate impulsive noises efficiently, and cannot perform well in the presence of signals that depends on noise. It is also known that for the precise properties of our visual system are not well understood this shows that our visual system requires nonlinear property.

3. DESIGN OF FILTER

In the present work a Gaussian pulse-shaping FIR filter has been designed and influence of its parameter variation has been investigated. The Gaussian pulse-shaping FIR filter has been designed by truncating the sampled version of a continuous-time impulse response of the Gaussian filter. The truncation and sampling introduce truncation error and sampling error in the design. The truncation error is due to FIR approximation of the IIR. The sampling error is occurs because a Gaussian frequency response is not band-limited.

Firstly, a continuous-time Gaussian filter has been designed by considering the symbol time, Ts = 1 micro second. The number of symbols between the start of the impulse response and its end has been taken as 6.

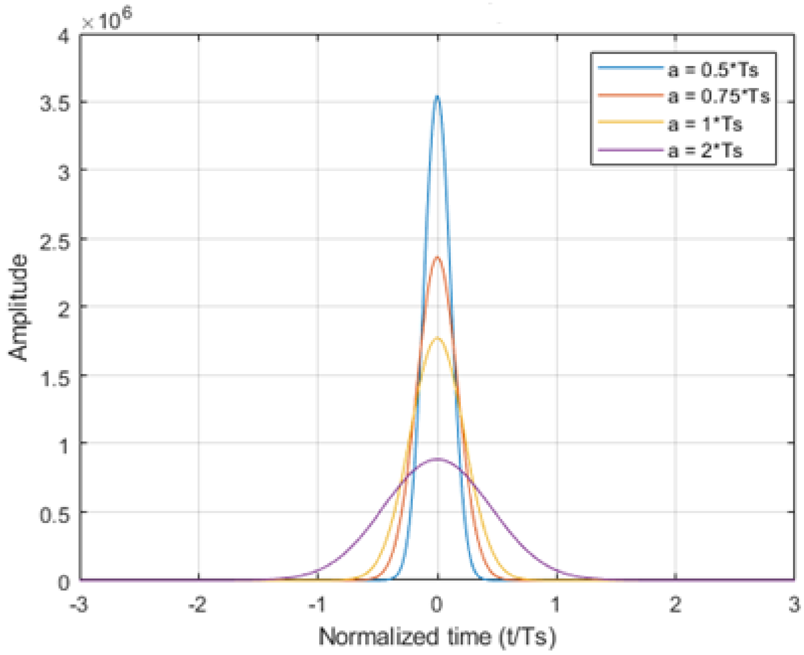

Figure 1: Impulse Response of the Continuous-Time Gaussian Filter for Various Bandwidths

To investigate the effect of the parameter ‘a’, the impulse response has been plotted for various values of ‘a’. Figure 1 shows the impulse response of the continuous-time Gaussian filter for various bandwidths. Figure 2 shows the ideal magnitude response of the continuous-time gaussian filter for various bandwidths.

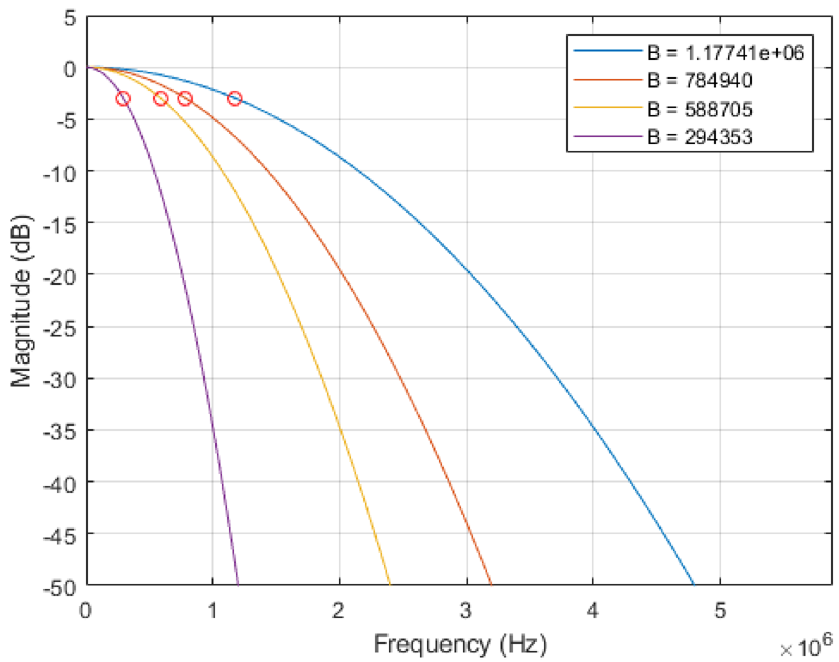

Figure 2: Ideal Magnitude Response of the Continuous-Time Gaussian Filter for Various Bandwidths

Then we have designed Gaussian pulse-shaping FIR filter by using approximation function. The 3-dB bandwidth symbol time product acts as inputs to this function and the number of symbol periods between the start and end of the filter impulse response is taken as 6. As the oversampling factor determines the sampling frequency and the filter length and hence this factor plays an important role in the Gaussian FIR filter design. The approximation errors in the design have been reduced by choosing appropriate value of oversampling factor and the oversampling factor has been taken as 16.

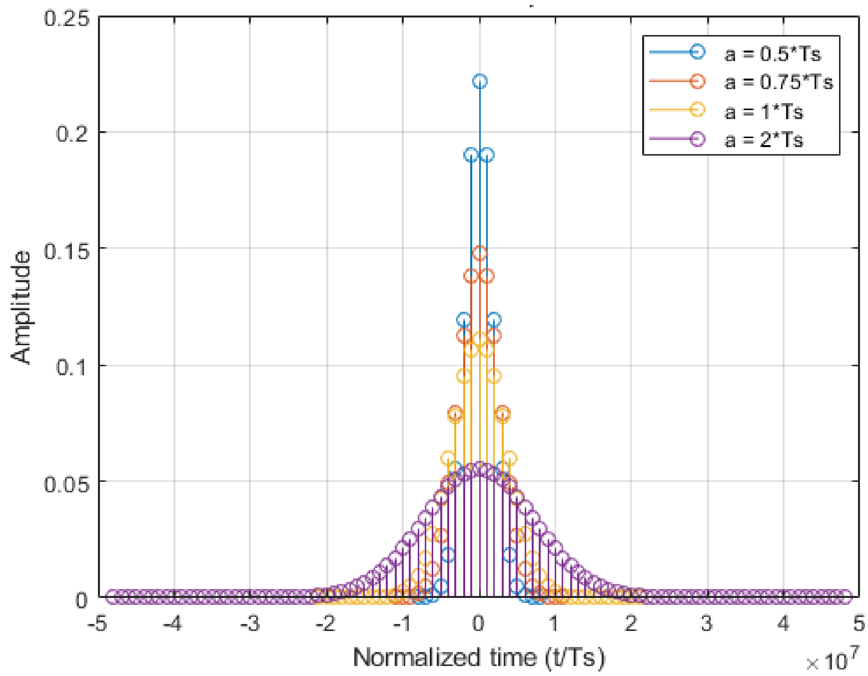

Figure 3: Impulse Response of the Gaussian Pulse-Shaping FIR Filter for Various Bandwidths for Oversampling Factor = 16

the frequency response for the Gaussian FIR filter with an oversampling factor of 16 has been calculated and has been compared it with the ideal frequency response shown in Figure 1 and Figure 3 impulse response of the gaussian pulse-shaping FIR filter for various bandwidths for oversampling factor = 16 and Figure 4 shows the ideal magnitude response of the gaussian pulse-shaping FIR filter for various bandwidths for Oversampling Factor = 16.

4. CONCLUSION

The Gaussian pulse-shaping FIR filter has been designed by truncating the sampled version of a continuous-time impulse response of the Gaussian filter. The truncation and sampling introduce truncation error and sampling error in the design. FIR filters suffer from aliasing errors, due to the aliasing occurs when the sampling frequency is not greater than the Nyquist frequency. In future the design of Gaussian pulse-shaping FIR filter could be design by using optimization techniques these techniques.

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